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4 Essential Requirements for VoIP Call Quality

Real-time services, like VoIP and video, place a different set of demands on a network than data. Provisioning the network to support these requirements ensures quality services. These are the four essential requirements every network must have, in priority order, to ensure VoIP call quality:

1) **Low Packet Loss.** If the network drops packets anywhere between the sender and receiver, information is lost and cannot be retransmitted. You can remedy a few lost packets in a conversation by selecting a codec that handles high loss environments. However, if the loss rate is too high, then clipping and dropped words will be encountered. If the loss rate goes even higher, half of the call may drop, creating one-way audio, or the entire call may drop.

2) **No Out-of-order packets.** Equally important, if packets arrive out-of-order, they cannot be re-sequenced back into the conversation. Out-of-order packets can occur when there are multiple paths to the destination, and the wrong load balancing mechanism is employed. For example: If two-gigabit links are trunked together to create a large, fault-tolerant pipe between two switches, and they are configured for per-packet load balancing, then packets for a single VoIP conversation may take either trunk link to reach its destination. Instead, the trunk port should be configured for per-flow load balancing to prevent this possibility. To a user, out-of-order packets would appear as packet loss: clipping, dropped words, one-way audio, and dropped calls.

3) **Low Jitter.** If some packets arrive early, and others later, the codec will do its best to re-assemble the conversation to flow naturally. But if the packets have too wide of a jitter range, there is only so much the codec can do. Some codecs may have a jitter buffer that can be adjusted to attempt to alleviate this situation. To ensure that jitter is low in your environment, track router CPU utilization and interface utilization and buffer error counters for all devices and links that pass VoIP traffic.

4) **Low Latency.** Latency is typically incurred due to the physical distance a packet travels. This can be low or high depending on how optimized the route is to the destination. For example: A coast-to-coast T1 circuit might have 60ms latency. Sending and receiving through a satellite is a significant additional distance to travel, and might be more than 600ms. Use route optimization and validation to ensure latency never gets out of control.

PathSolutions TotalView covers all of these bases, ensuring the best operating environment for VoIP and video services.

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